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H04Q 1/444

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H4K KAV

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(54) Abstract Title
Inband signalling

(57) A speech signal and data are transmitted simultaneously in different portions of the speech band, and the bandwidth of the speech portion is controlled according to the amount of data being transmitted. Preferably the high-frequency cut-off point for the speech signal is reduced to accommodate the data. The data can be out-of-band signalling data which is converted to in-band data so as to fit within the restricted bandwidth of a multiplexed channel.

Fig.6A.

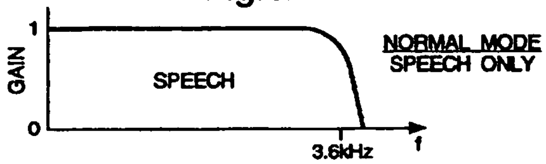


Fig.6B.

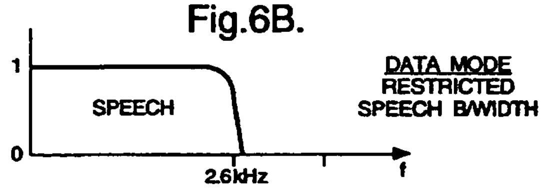


Fig.6C.

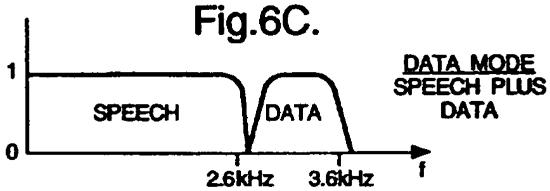
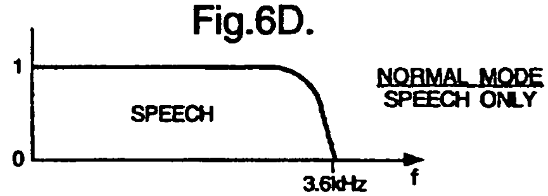


Fig.6D.



At least one drawing originally filed was informal and the print reproduced here is taken from a later filed formal copy.

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Fig.1.

(PRIOR ART)

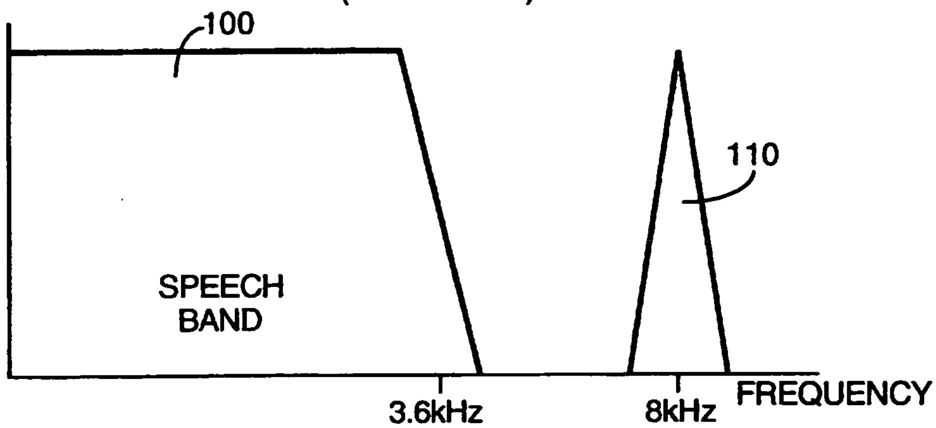


Fig.2.

(PRIOR ART)

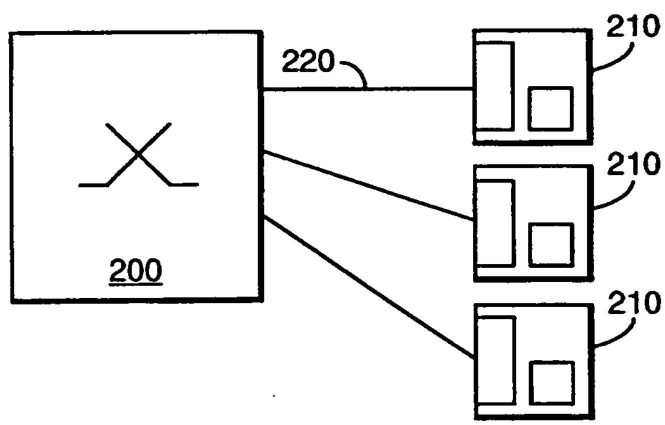
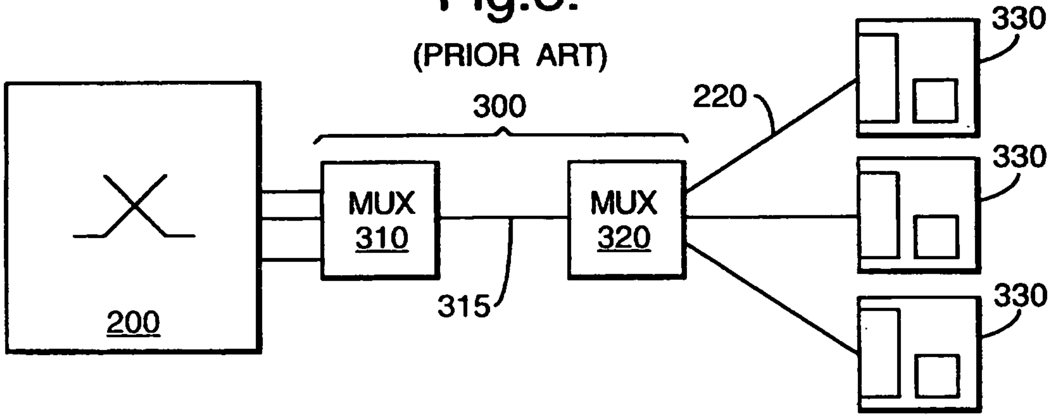


Fig.3.

(PRIOR ART)



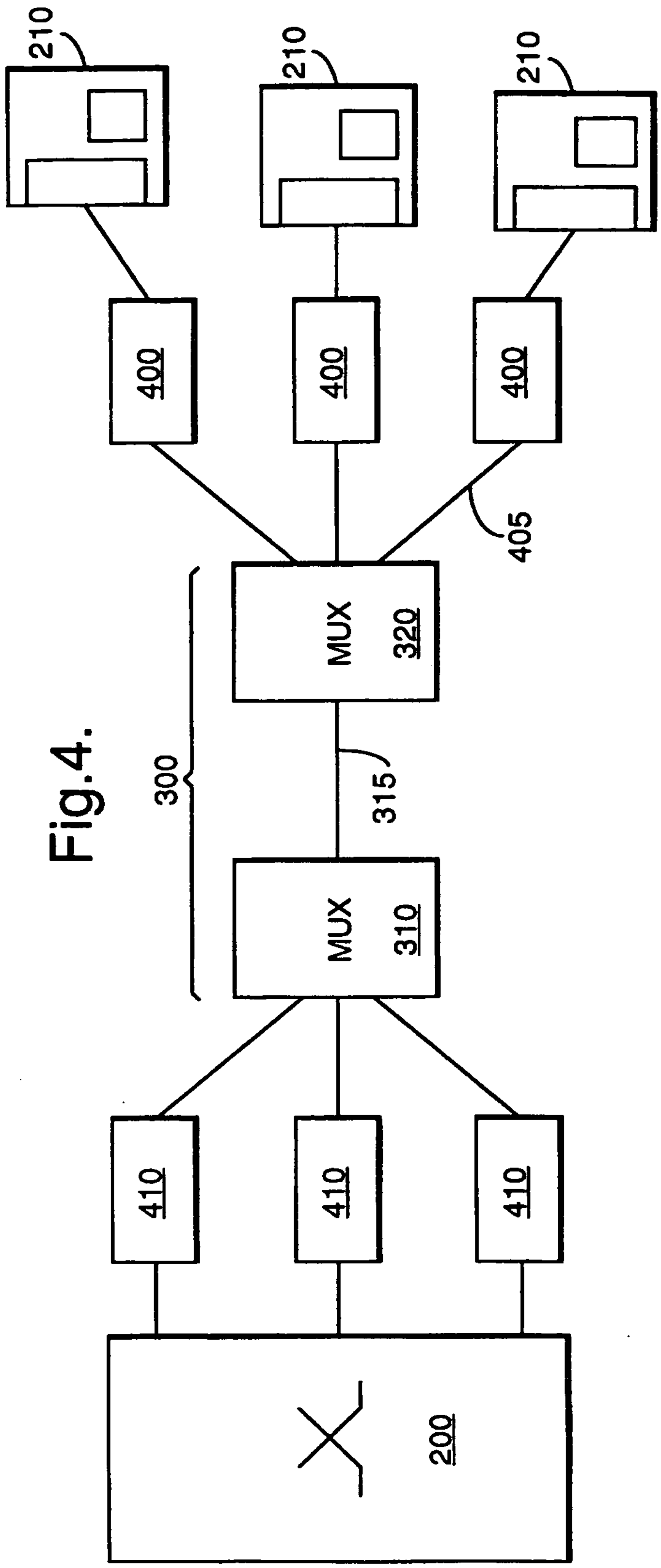

210
210
210
400
400
400
405
MUX
320
300
315
MUX
310
410
410
410
200

Fig.4.

Fig.5.

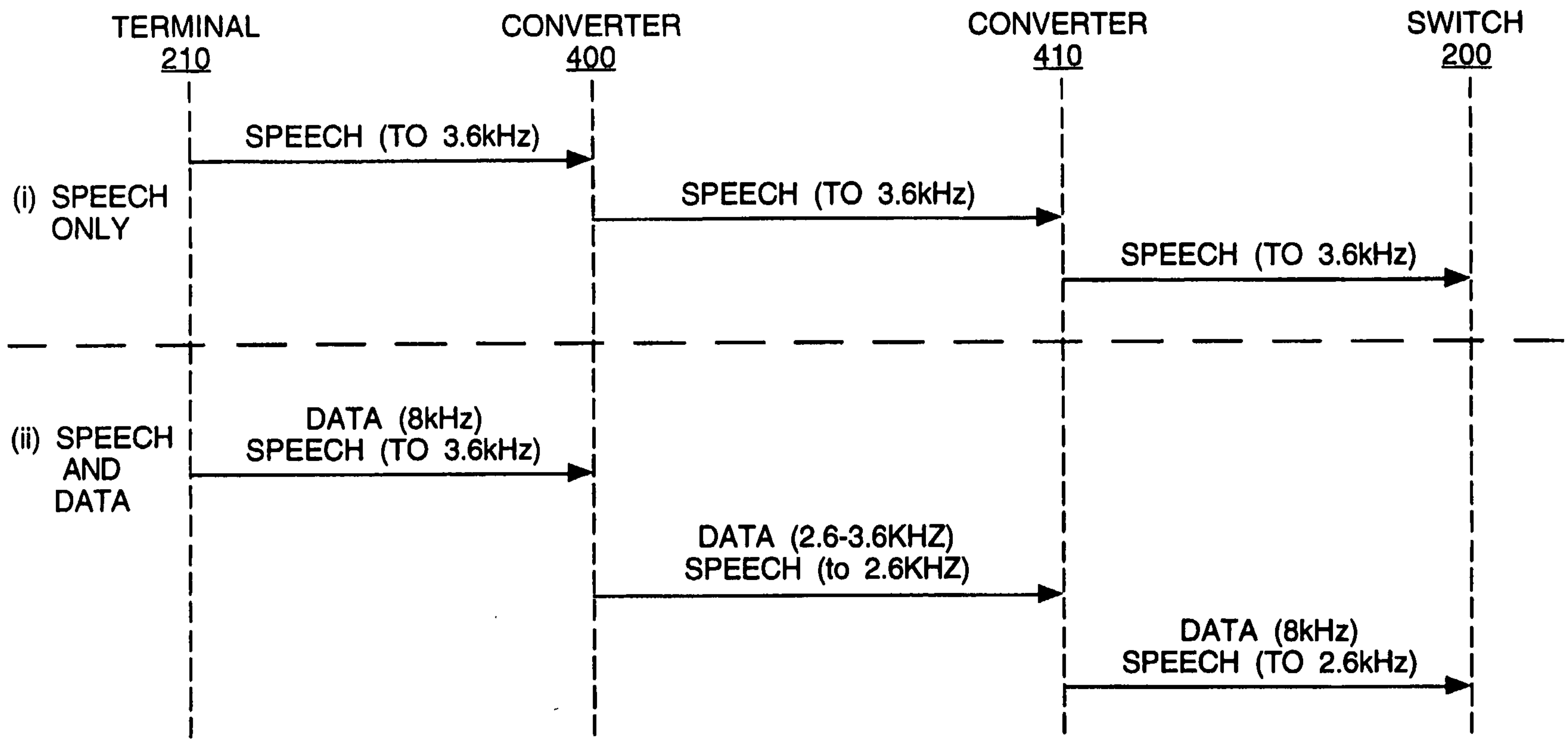

TERMINAL
210
CONVERTER
400
CONVERTER
410
SWITCH
200
SPEECH (TO 3.6kHz)
SPEECH (TO 3.6kHz)
SPEECH (TO 3.6kHz)
(i) SPEECH ONLY
(ii) SPEECH AND DATA
DATA (8kHz)
SPEECH (TO 3.6kHz)
DATA (2.6-3.6KHZ)
SPEECH (to 2.6KHZ)
DATA (8kHz)
SPEECH (TO 2.6kHz)

Fig.6A.

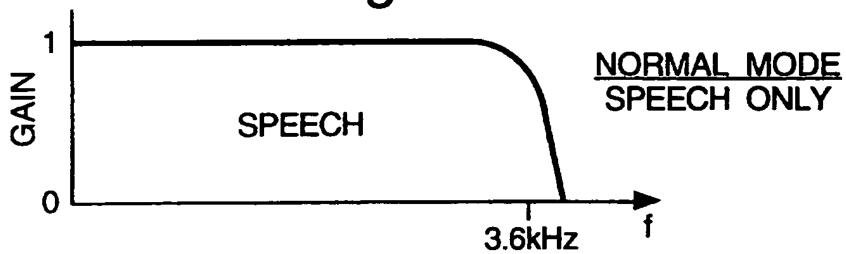


Fig.6B.

Fig.6C.

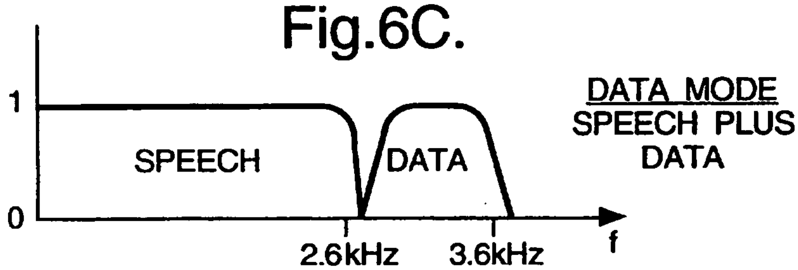


Fig.6D.

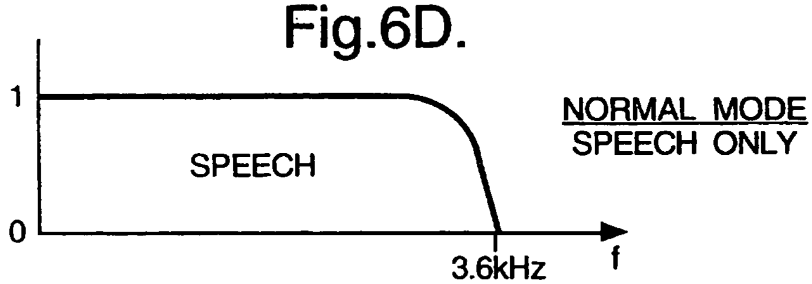


Fig.7.

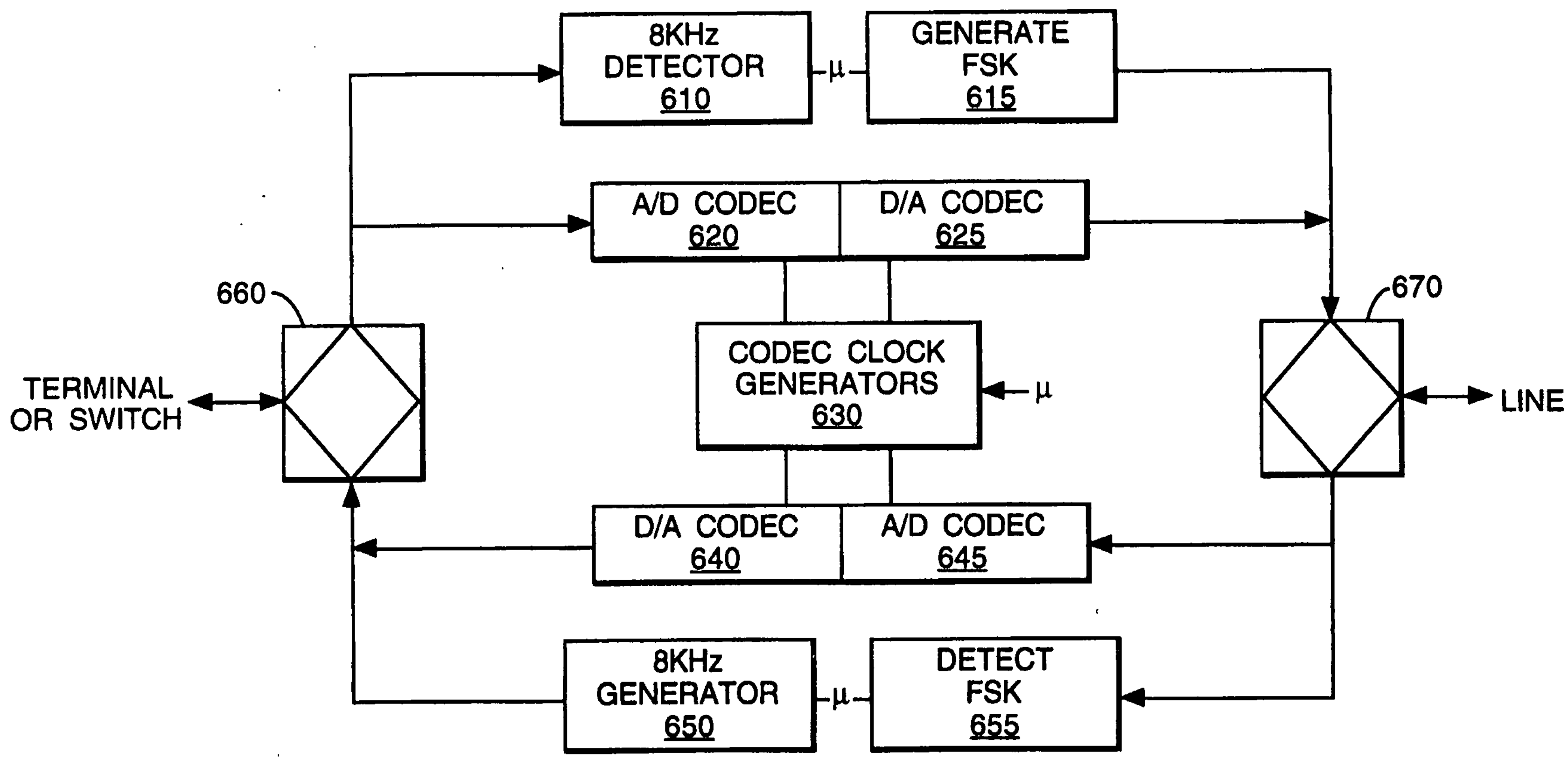

8KHz
DETECTOR
610
μ
GENERATE
FSK
615
A/D CODEC
620
D/A CODEC
625
660
670
TERMINAL
OR SWITCH
CODEC CLOCK
GENERATORS
630
μ
LINE
D/A CODEC
640
A/D CODEC
645
8KHz
GENERATOR
650
μ
DETECT
FSK
655

Fig.8.

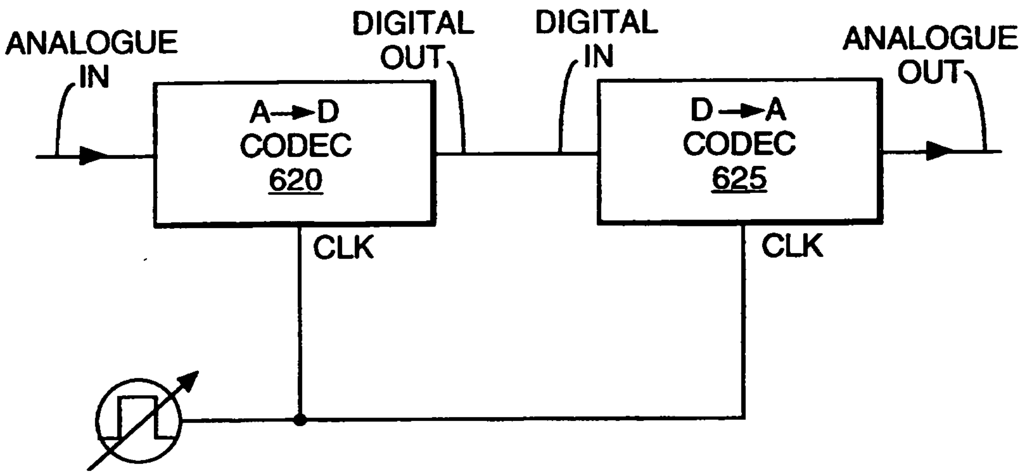

ANALOGUE
IN
DIGITAL
OUT
DIGITAL
IN
ANALOGUE
OUT
A→D
CODEC
620
D→A
CODEC
625
CLK
CLK

Fig.9.

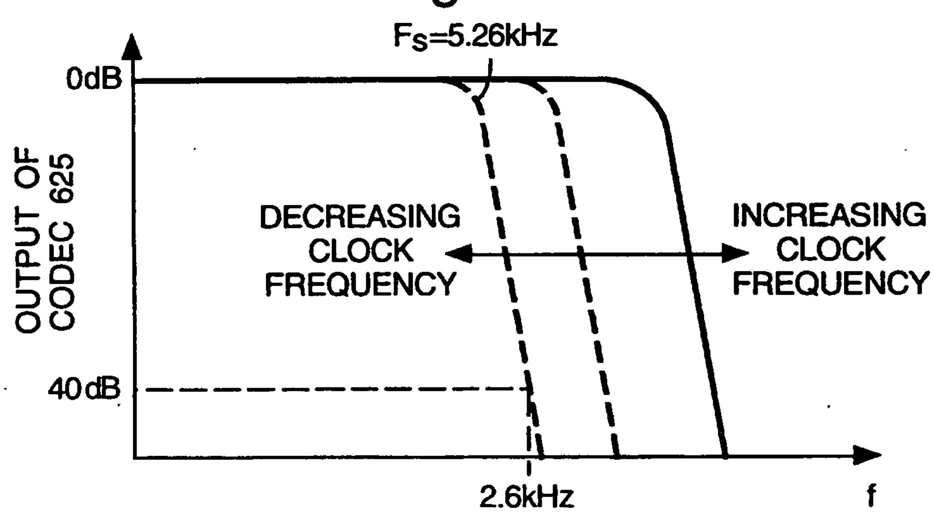

Fs=5.26kHz
0dB
OUTPUT OF
CODEC 625
DECREASING
CLOCK
FREQUENCY
INCREASING
CLOCK
FREQUENCY
40dB
2.6kHz
f

Fig.10

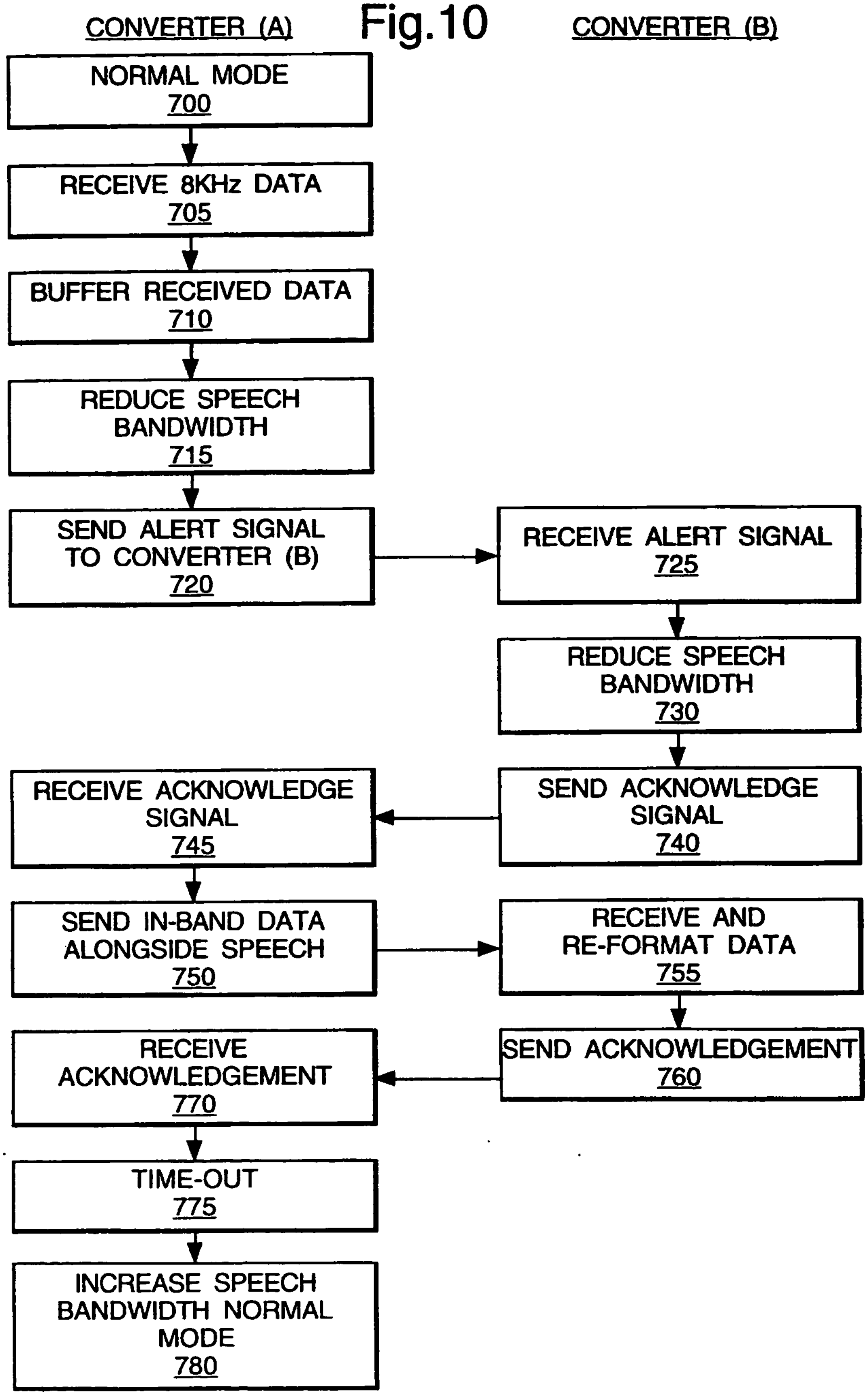

CONVERTER (A)
CONVERTER (B)
NORMAL MODE 700
RECEIVE 8KHz DATA 705
BUFFER RECEIVED DATA 710
REDUCE SPEECH BANDWIDTH 715
SEND ALERT SIGNAL TO CONVERTER (B) 720
RECEIVE ALERT SIGNAL 725
REDUCE SPEECH BANDWIDTH 730
SEND ACKNOWLEDGE SIGNAL 740
RECEIVE ACKNOWLEDGE SIGNAL 745
SEND IN-BAND DATA ALONGSIDE SPEECH 750
RECEIVE AND RE-FORMAT DATA 755
SEND ACKNOWLEDGEMENT 760
RECEIVE ACKNOWLEDGEMENT 770
TIME-OUT 775
INCREASE SPEECH BANDWIDTH NORMAL MODE 780

Fig.11

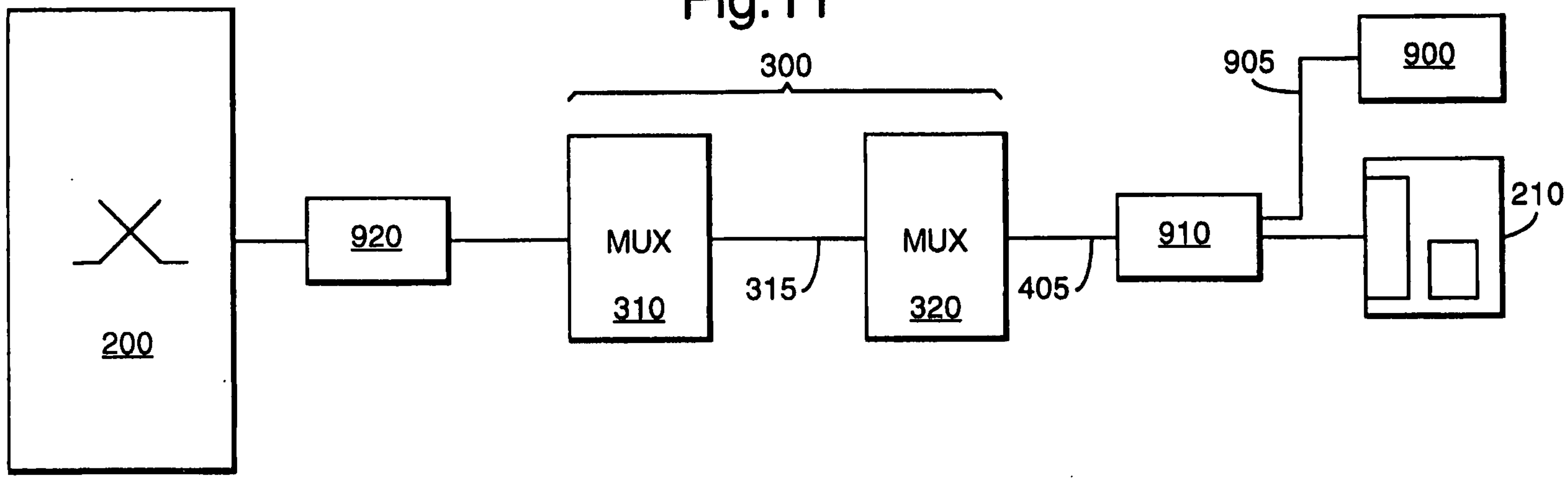
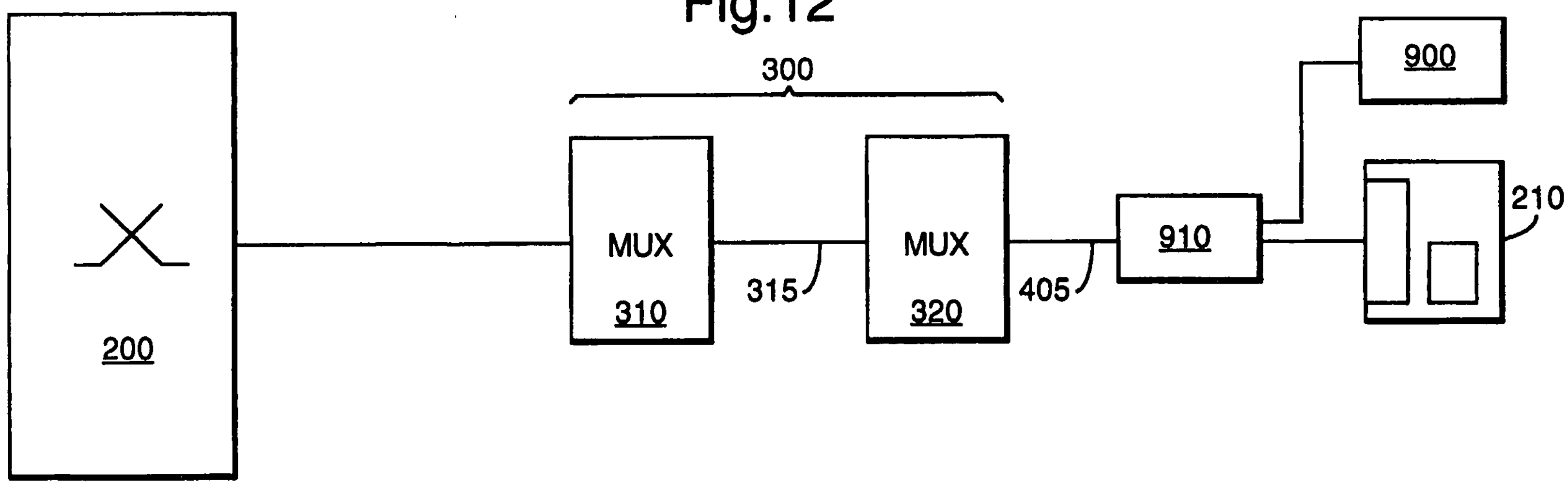

200
920
300
MUX
310
315
MUX
320
405
910
905
900
210

Fig.12


200
300
MUX
310
315
MUX
320
405
910
900
210

2321822

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- T LEWIS - J. HEARFIELD 8-4-1-1 US

METHOD AND APPARATUS FOR CONVEYING DATA OVER A COMMUNICATIONS CHANNEL

TECHNICAL FIELD

This invention relates to conveying a speech signal and data, such as telecommunications signalling data, along a speech band of a communications channel.

BACKGROUND OF THE INVENTION

In the local loop of a Public Switched Telephone Network (PSTN) signalling data is sent between a switch and a subscriber's terminal apparatus to perform such tasks as call set-up, calling line identification and call clearing. One known method of conveying signalling data between a switch and a terminal is by the loop-disconnect method, where the direct-current loop between the switch and terminal is modified in a pulse-like manner. This method enables simple information to be carried but has a disadvantage of being slow. Another known method of conveying signalling data is by multi-frequency (MF) signalling. This method uses various combinations of two speech-band tones which are transmitted simultaneously. This method offers a faster transfer of signalling information.

A further known method of conveying signalling data is by out-of-band signalling, where signalling between a switch and a subscriber terminal occurs at a frequency lying outside the frequency band used for carrying speech. This method is illustrated in figure 1. In a system manufactured by Nortel called 'P-Phone' the speech bandwidth 100 is limited to about 3600Hz and out-of-band signalling is conveyed by an 8kHz carrier tone 110, which is amplitude-shift keyed with the signalling data.

Out-of-band signalling offers a variety of benefits over either loop disconnect or multi-frequency signalling. The main benefit is that during speech transmission a user does not have their conversation disrupted by signalling of information between the terminal and the switch, because the data signalling occurs outside the speech band.

Usually terminals 210 using out-of-band signalling are each connected directly by a twisted-pair cable 220 to a switch 200 as shown in figure 2. This arrangement is practical for a typical office environment where a building has a switch 200 which is dedicated to serving terminals 210 within that building. However, there is a limit of around 5km on the length of line 220 which is allowed between a switch and a terminal. Errors start to occur on line lengths greater than this, due to the attenuation effect of transmission lines. It is also uneconomical to use physically separate lines to couple each terminal to a switch where the terminals are separated from the switch by a large distance.

In the event that terminals which use out-of-band signalling are required at line lengths greater than 5Km, remote units are used. Remote units multiplex a group of terminal voice channels and a group of out-of-band signalling channels. These units are expensive and require substantial local space. It is therefore desirable to avoid using remote units.

A conventional method of connecting a group of subscriber terminals to a remote switch is shown in figure 3. A group of terminals 330 are connected to a multiplexer 320, with a trunk line 315 coupling between multiplexer 320 located at a customer site or local exchange, and a further multiplexer 310 which is usually located at the site of the remote switch 200. This arrangement works well with terminals employing conventional in-band signalling. All commercially available multiplexers comply to standards which limits the bandwidth of the signal received on channel 220 to about 3.6kHz and samples the signal to generate a 64Kbit/s data stream for each channel. This is conveyed, in digital form, over trunk 315 in a time-multiplexed manner

along with the 64kbit/s data streams for other channels 220, as is well-known in the art. If a terminal employing out-of-band signalling were connected to such a multiplexer, the out-of-band data would be removed by the filtering operation at the multiplexer 320 and would not be received by the switch 200. Similarly, in the direction of transmission from switch 200 to terminal 330, out-of-band signalling would be removed by the filtering operation at multiplexer 310.

One solution to this problem is proposed in UK Patent Application GB9604381.5, where a first signalling converter is installed at the switch and a second signalling converter is installed at the terminal apparatus. The signalling converters are arranged to receive out-of-band signalling messages and to convert them to messages lying within the speech band such that they can be carried within the restricted bandwidth of the multiplexed network 300.

Issued Patent US 4,010,328 (McGuire) describes an apparatus for adapting payphones to carrier telephone systems where dc payphone signalling is conveyed over the carrier system in a part of the speech band.

The present invention seeks to provide an improved method and apparatus for conveying data over a communications channel.

SUMMARY OF THE INVENTION

According to the present invention there is provided a method of transmitting a speech signal and data along a speech band of a communications channel, the method comprising:

- transmitting the speech signal and data simultaneously in different portions of the speech band, and
- controlling the bandwidth of the speech portion according to the amount of data being transmitted.

Another aspect of the invention provides apparatus for use in transmitting a speech signal and data along a speech band of a

communications channel, the converter being coupled to the channel and comprising:

- means for transmitting the speech signal and data simultaneously in different portions of the speech band, and
- means for controlling the bandwidth of the speech portion according to the amount of data being transmitted.

A further aspect of the invention provides a method of receiving a speech signal and data which are transmitted in the above manner, which method comprises the steps of:

- determining an amount of data being transmitted, and
- separating the data from the speech signal according to the determination.

A still further aspect of the invention provides apparatus for receiving a speech signal and data according to the above method.

An advantage of fitting the signalling data within a portion of the channel bandwidth which is separate from that used for conveying speech is that data is not heard by a subscriber. In fact, the subscriber could be unaware that data is being transferred. Data is maintained separate from the speech signal at all times so that speech and data cannot corrupt each other. By controlling the allocation of bandwidth dynamically, in response to the amount of data (if any) which needs to be sent, the speech portion is restricted in bandwidth only for as long as is necessary to send data.

The bandwidth of the speech portion can be controlled by controlling the high-frequency cut-off point of the speech signal.

The data can comprise signalling data, such as signalling data which relates to a telecommunications call of which the speech signal is a part.

The channel can comprise a restricted bandwidth section of a network, with the method being performed so as to convey a speech signal and

data over the restricted bandwidth section. This is particularly useful where the data is conveyed outside the speech band at other parts of the network away from the restricted bandwidth section, the speech and data together occupying a bandwidth which is greater than that provided by the restricted bandwidth section.

One application of this arrangement is in systems where a telecommunications switch and subscriber terminal convey signalling data via an out-of-band tone, but where there is a need to have the switch and terminal communicate via a channel having a restricted bandwidth, such as a multiplexed 3.6kHz channel, which is too narrow to accommodate the out-of-band tone.

The apparatus is not restricted to use with telephone signalling data and can be used where there is a need to convey any kind of data over a communications channel alongside speech. Telemetry and subscriber meter reading data are examples of such applications.

Preferably information relating to the controlled bandwidth is conveyed to a receiving end of the channel. This information helps the receiving end to separate the speech and data. In the case where the channel carries (i) just speech, or (ii) speech plus a predetermined data bandwidth, the information can simply comprise an alert signal that data is to be sent, which the receiving end can use to set predetermined speech and data bandwidths.

The step of controlling bandwidth of the speech portion of the speech band can comprise passing the speech signal sequentially through the analogue-to-digital and digital-to-analogue parts of codecs and reducing the bandwidth of the speech by varying the filtering characteristics of the codecs. This can be achieved by varying clock rate at which the codecs operate. The advantages of using codecs are that they are low-cost, reliable, have a low parts count and a sharp filter response which is easily controlled.

BRIEF DESCRIPTION OF THE DRAWINGS

For a better understanding of the invention, and to show by way of example how it may be carried into effect, embodiments will now be described with reference to the accompanying drawings in which:

Figure 1 shows the allocation of bandwidth in a known signalling system which uses an out-of-band tone;

Figure 2 shows a group of terminals, each connected directly to a switch in a known manner;

Figure 3 shows a group of terminals which are connected to a switch by a multiplexed trunk in a known manner;

Figure 4 shows an arrangement in which terminals using out-of-band signalling are connected to a switch by a multiplexed trunk;

Figure 5 shows how speech and data are carried over the arrangement of figure 4;

Figures 6A to 6D show how a data converter allocates bandwidth in operation;

Figure 7 shows a data converter of figure 4 in more detail;

Figure 8 shows the codecs of figure 7 in more detail;

Figure 9 shows the performance of the codecs of figure 8;

Figure 10 is a flow diagram showing the operation of a pair of data converters;

Figure 11 shows an alternative arrangement to that of figure 4 in which a data source is separate from a terminal;

Figure 12 shows an alternative arrangement to that of figure 11.

DESCRIPTION OF PREFERRED EMBODIMENTS

Figure 4 illustrates an embodiment of the present invention. Unlike figure 3, where the terminal and the switch are connected directly to the multiplexed trunk of restricted bandwidth, a first data converter 400 is coupled between terminal 210 and multiplexer 320, and a second data converter 410 is coupled between the switch-end multiplexer 310 and the switch 200. The data converters 400, 410 receive out-of-band signalling from either the switch 200 or the terminal 210 and convert this to an in-band signal which can be sent over the bandwidth

restricted network 300. After transmission over the bandwidth restricted network 300 the in-band signalling is received by a second data converter and converted back to the out-of-band signalling format that the switch or terminal can recognise. Data conversion can occur in either direction. Data converters 400, 410 reduce the allocation of channel bandwidth to speech for the time that data is transmitted in order to accommodate the data signal. However, to minimise effect that the data channel has on the speech channel, it is preferred to restrict data transmission to only one direction at a time. The signalling data is used to support a range of PABX features such as display of calling number, display of time and date, call redirection.

Figure 5 shows in more detail how (i) speech only, and (ii) speech plus data are carried between terminal 210 and switch 200 of figure 4. In the speech only case, speech is sent from terminal 210 occupying the band up to 3.6kHz. The speech is received at converter 400 and is conveyed to converter 410 occupying the full 3.6kHz available on multiplexed network 300. Finally, the speech is sent to switch 200. Thus, the speech occupies the band up to 3.6kHz for the full path between terminal 210 and switch 200, in both directions of transmission.

Part (ii) of figure 5 shows the case where terminal 210 transmits both speech and data. Terminal 210 sends speech (up to 3.6kHz) and data at 8kHz. Converter 400 receives the speech and narrows the speech to 2.6kHz, converts the 8kHz data to fit within the band 2.6kHz to 3.6kHz which is made available by the narrowing of the speech, and transmits both speech and data over the 3.6kHz channel available on multiplexed network 300. At converter 410 the data is converted back to out-of-band 8kHz data. The output of converter 410 is speech having a reduced bandwidth (2.6kHz) and out-of-band 8kHz data. The in-band data which occupies the 2.6 to 3.6kHz band of the multiplexed network is suppressed by converter 410 so that an end-user is not distracted by it. Operation of the arrangement for transmitting signalling data in the reverse direction, from switch 200 to terminal 210, is similar to that described above.

Data converter 400 can be installed local to terminal 210, and can comprise a single card powered from the local mains supply. The data converter is provided with connections to couple it to the terminal and to line 405.

Data converter 410 associated with switch 200 can be provided as a single line card with connections for coupling to the switch 200, multiplexer 310 and local power. Preferably a number of data converters are located on a single card for fitting in a rack of multiplexer 310, or a rack of switch 200. These converters support data conversion for a group of separate lines, e.g. a group of twelve lines. Powering may be derived from the local mains supply, via a suitable mains transformer, or by a 50V feed derived from the host multiplexer 310 or switch 200. Connection to the switch and network can be via DIN 4162 type connections. Switch 200 can be a DMS® switch manufactured by Northern Telecom. Multiplexer 310 can be a PDMX® manufactured by Northern Telecom.

Figures 6A to 6D show the allocation of bandwidth on the channel between data converters 400, 410 in figure 4 during operation of the system.

Figure 6A shows the allocation in 'normal mode', where speech, in analogue form, occupies the usual bandwidth up to 3.6kHz, which is the cut-off frequency of filters at multiplexers 310,320. This is the usual state in the absence of any data to send.

Figure 6B shows the allocation of bandwidth in 'data mode'. The speech bandwidth is narrowed from 3.6kHz down to 2.6kHz to leave a block of bandwidth to accommodate the data channel.

Figure 6C, following on from figure 6B, shows the allocation of bandwidth in 'data mode' with the restricted speech (0 - 2.6kHz) and a separate data band (2.6kHz to 3.6kHz) co-existing. Data and speech

can be simultaneously carried over a bandwidth-restricted channel without interference between the speech and data.

Figure 6D is the same as figure 6A, and shows the allocation of bandwidth in 'normal mode' following transmission of data. The allocation of bandwidth to speech has been restored to 3.6kHz.

The data converters 400, 410 located at the terminal and switch ends are similar in operation. It is preferred to construct one type of converter which is provided with a set of option links. The option links on the converter control the operation of the converter as a 'terminal-based' or 'switch-based' converter. These option links are set to the required positions at the time of manufacture or installation. Figure 7 shows one such data converter in more detail.

Operation of the converter is controlled by a microcontroller μ, which is fed by a 14.3MHz clock. The converter has an 8kHz detector 610 and an 8kHz generator 650 to respectively detect and generate out-of-band signalling. 8kHz detector 610 detects and decodes out-of-band signalling received on a line from terminal 210 or switch 200. The output of the detector is a digital serial bit stream which is fed to microcontroller μ. The 8kHz generator 650 consists of an 8KHz sine wave generator which is modulated with serial data from microcontroller μ to generate out-of-band signalling data which can be read by the terminal or switch. In Northern Telecom's 'P-Phone' signalling data is carried by an 8kHz tone which is Amplitude Shift Keyed (ASK) with the data at a rate of 1kbit/s. Signalling messages are 16-bits long, including start and end flags and a parity bit.

The converter also has an FSK generator 615 and FSK detector 655 to support the in-band data channel. FSK generator 615 has two functions. Firstly, it is used to generate the in-band FSK data. It generates FSK data with signalling components at 2.7 and 3.4kHz. The FSK generator consists of a sine wave generator, the frequency of the sine wave being controlled by microcontroller μ. The maximum data rate is 500bps with a signal amplitude set at -15dBV. Secondly,

the FSK generator is used to generate an alert tone for establishing the 'data mode'. For this function the generator is set to generate a fixed-frequency tone 2.7KHz for a specific time period.

The FSK detector 655 is used both to detect the FSK data signalling components at 2.7 and 3.4kHz and to detect the alert tone signalling component at 2.7KHz. FSK detector 655 can use tone-decoder band-pass filters or phase-locked loops. Conveying signalling data over the data channel by the use of two tones at 2.7kHz and 3.4kHz is preferred. However, pairs of tones at different frequencies, or a different number of tones could be fitted within the data bandwidth to convey data. It is also possible to use modulation schemes other than FSK, such as quadrature phase-shift keying (QPSK), to convey data. Such schemes allow a higher data throughput in the limited bandwidth of the data channel.

Speech signals passing through the data converter are routed through a low-pass filter, whose upper cut-off frequency is controlled by the microcontroller. The cut-off frequency of the low-pass filter depends on whether just speech signals or speech signals plus signalling data are carried over the channel.

The low-pass filter can be realised by chaining together the analogue-to-digital and digital-to-analogue functions of a codec, as shown by 620, 625 and 640, 645 in figure 7. The A-to-D and D-to-A functions of a single or two different codecs can be chained together.

Figure 8 shows the codecs in more detail. Speech signals are fed into the analogue input of the A-to-D part 620 of a codec and are sampled. The sampling rate is set by the rate of the clock signal CLK which is fed to the codec. In normal operation, where a speech bandwidth of 3.6kHz is required, codec 620 samples at a rate of 8KHz. Because of additional analogue filtering within the codec the actual value is a nominal 3.6KHz. The digital sampled signal is coupled to the digital input of the D-to-A part 625 of a codec, which converts the sampled

signal back to an analogue signal at a rate which is governed by clock signal CLK.

By Nyquist's sampling theory it can be seen that the reconstituted signals at the analogue output of 625 will be band-limited to around one half of the sampling frequency. By reducing the clock rates fed to the pair of codecs, and hence the sampling rate at which the codecs operate, it can be seen that the pass-band of the codecs becomes narrower. Each chained pair of the A-to-D and D-to-A functions of a codec thus acts as a programmable low-pass filter with a high-Q response, i.e. a sharp roll-off, the cut-off frequency being a direct function of the clock frequency which is fed to the codecs. Clock signals are generated by generator 630 under the control of the microcontroller.

Figure 9 shows how the filter response of a codec changes with a change in clock frequency. It has been found that in 'normal mode', with a speech-only signal, codec 620 requires it's normal 8kHz sampling rate. For 'data mode' where a narrower speech bandwidth is required, the sampling rate is reduced to 5.26kHz, giving a 40dB attenuation at 2.6kHz. This allows data to be carried in the band 2.6 - 3.6kHz without corruption occurring between the speech and data.

Hybrids circuits 660, 670 act as 2-to-4 wire converters. A dc blocking capacitor and a 1:1 matching transformer (not shown) couple a hybrid to a telephone line or terminal.

The operation of a pair of data converters A and B will now be described with reference to the flow diagram shown in figure 10.

In normal mode 700 the clocking speed of the codec blocks 620,625 and 640,645 is set to give a bandwidth of 3.6kHz for the voice path between converters 400 and 410. Because the bandwidth of the multiplexed network 300 is also 3.6kHz in this configuration speech transmission between terminal and switch is as it would be without data converters in the network, and speech performance is unimpaired.

In step 705 out-of-band 8kHz data is received at converter A. This is stored in a buffer at converter A at step 710. Receiving out-of-band data is a trigger for converter A to proceed to step 715 where data converter A reduces the clock speed to the codecs to give a bandwidth for speech of 2.6kHz. At step 720 data converter A sends an alert signal to converter B indicating that it is ready to send data using the FSK generator block 615. The alert signal is a 2.7KHz tone which lasts for 500mS.

On receiving the alert signal, in step 725, converter B reduces the speed of its data clock 630 to give a reduced bandwidth for speech of 2.6kHz, at step 730. At step 740 data converter B then sends an acknowledge signal using the FSK generator block 615 to converter A. The acknowledge signal is a 12 bit FSK data word.

Figures 6B and 6C, as previously described, show the allocation of bandwidth at this time. Clean and separate speech and data channels now exist.

When converter A receives the acknowledge signal from converter B, at step 745, it then transmits (step 750) the buffered data to converter B. Data is reformatted into 12 bit words for transmission in-band using the FSK generator 615. When converter B receives the in-band data it reformats it (step 755) and transmits it to the switch or terminal in 8kHz out-of-band format. The output of converter B thus comprises speech at a reduced bandwidth (up to 2.6kHz) and out-of-band data (at 8kHz). The in-band FSK data is suppressed by converter B such that it is not received by a user.

Because of potential network delays of several ms, to monitor channel status, and also allow for error prevention, after receiving and decoding a valid in-band message the data converter B transmits an acknowledgement to converter A, at step 760, to confirm it has received the message.

The acknowledge signal is a 12bit FSK data word. In the event that converter A does not receive the valid acknowledge signal within a given time-out converter A retransmits the message a second or third time. This allows for basic error prevention and channel monitoring. In the event that it does not receive a valid acknowledge after the third transmission the data is lost. Other forms of handshaking could be used as an alternative to this.

Additional incoming out-of-band messages received by converter A are buffered until a valid acknowledge signal is received from converter B. The additional messages are then transmitted in-band to converter B as in step 750. Additional incoming out-of-band messages received by converter B are buffered until a channel time-out has occurred. Following the time-out converter B transmits the buffered messages to converter A and takes over control of the data channel.

After completion of data transfer each of the data converters maintains a separate data channel for a time-out period (step 775). The converters then independently return to the idle state and return the codec clocking speeds to allocate the full 3.6kHz channel bandwidth to speech (step 780).

Data transmissions in the opposite direction i.e. from converter B to converter A, are processed in a similar manner to that just described.

In the embodiments just described it is signalling data from a terminal which is conveyed within the data band. Figures 11 and 12 show a further application of the equipment. In this application it is data from a data source 900 which is conveyed within the data band. A data converter 910 is coupled to a telephone 210, as before, and also a data source 900 via a line 905. The data source can be a telemetry unit, meter reading device or some other source of data. Data converter 910 operates as before, in that:

(i) when only speech needs to be sent, it conveys the speech over channel 405 using the full available bandwidth of multiplexed network 300;

(ii) when speech and data need to be sent, it narrows the bandwidth allocated to speech to allow the data to be carried within a separate data band within the bandwidth of multiplexed network 300.

Data converter 910 may itself sense that data needs to be carried over channel 405, by the act of receiving data over line 905. Alternatively, it may respond to an instruction which is included in the data that it receives over line 905.

Data converter 910 allocates bandwidth in the same manner as shown in figures 6A to 6D. The functional blocks of the converter 910, local to the subscriber, are modified from those shown in figure 7, by replacing the 8kHz detector 610 and 8kHz generator 655 by a respective detector and generator which are compatible with the modulation and encoding scheme used by data source 900. A data converter 920 at the switch end of the system converts the data into a format which the switch can receive.

In the alternative arrangement of figure 12, the switch-end data converter 920 of figure 11 is omitted, and the data remains in the data band through to it's destination. The destination terminal apparatus (not shown) incorporates a data converter which removes the data from the data band and converts it into a form which is interpretable by the destination terminal apparatus.

In the preceding description the data converter has been described as a unit which is separate from the subscriber terminal. It is also possible to incorporate the converter into the communications terminal, with the converter being controlled by a microcontroller within the terminal apparatus. In this embodiment the speed of data transfer is increased because of the removal of the need to decode out-of-band messages.

CLAIMS

1. A method of transmitting a speech signal and data along a speech band of a communications channel, the method comprising:

- transmitting the speech signal and data simultaneously in different portions of the speech band, and

- controlling the bandwidth of the speech portion according to the amount of data being transmitted.

2. A method according to claim 1 wherein when no data is transmitted the bandwidth of the speech portion is controlled to fill the speech band.

3. A method according to claim 1 wherein the step of controlling the bandwidth of the speech portion comprises controlling the high-frequency cut-off point of the speech signal.

4. A method according to claim 1 wherein the step of controlling the bandwidth is carried out when the speech is in the form of an analogue signal.

5. A method according to claim 1 wherein the data comprises signalling data.

6. A method according to claim 5 wherein the signalling data relates to a call of which the speech signal is a part.

7. A method according to claim 1 further comprising determining that data needs to be transmitted over the channel, and controlling the bandwidth of the speech in response to that determination.

8. A method according to claim 1 wherein information relating to the controlled bandwidth is conveyed to a receiving end of the channel.

9. A method according to claim 1 wherein at least part of the channel comprises a restricted bandwidth section of a network, and the

method is performed so as to convey a speech signal and data over the restricted bandwidth section.

10. A method according to claim 9 wherein the data is conveyed outside the speech band at other parts of the network away from the restricted bandwidth section.

11. A method according to claim 1 comprising passing the speech sequentially through the analogue-to-digital and digital-to-analogue parts of codecs and controlling the bandwidth of the speech portion by varying the filtering characteristics of the codecs.

12. A method according to claim 11 wherein the filtering characteristics of the codecs are varied by varying clock rate at which the codecs operate.

13. A method of receiving a speech signal and data which are transmitted according to a method of claim 1 comprising the steps of:

- determining an amount of data being transmitted, and
- separating the data from the speech signal according to the determination.

14. Apparatus for use in transmitting a speech signal and data along a speech band of a communications channel, the converter being coupled to the channel and comprising:

- means for transmitting the speech signal and data simultaneously in different portions of the speech band, and
- means for controlling the bandwidth of the speech portion according to the amount of data being transmitted.

15. Apparatus according to claim 14 wherein the apparatus is incorporated in a communications terminal.

16. A telecommunications network incorporating an apparatus according to claim 14.

17. Apparatus for receiving a speech signal and data which are transmitted according to a method of claim 1 comprising:

- means for determining an amount of data being transmitted, and

- means for separating the data from the speech signal according to the determination.

18. A method of filtering an analogue signal by passing it sequentially through the analogue-to-digital and digital-to-analogue parts of codecs,
and selecting the filtering characteristics by varying clock rate at which the codecs operate.

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Databases searched:

UK Patent Office collections, including GB, EP, WO & US patent specifications, in:

UK Cl (Ed.O): H4K: KAE; KAM; KAV

Int Cl (Ed.6): H04M, H04Q

Other:

Documents considered to be relevant:

Category	Identity of document and relevant passage	Relevant to claims
A	GB 2 310 778 A (NORTHERN)	
A	US 5 353 342 (PIETROWICZ)	

X	Document indicating lack of novelty or inventive step	A	Document indicating technological background and/or state of the art.
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